NONPROVISIONAL PATENT APPLICATION

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

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Attorney Docket No.: 103176

Date: April 1, 1999

BOX PATENT APPLICATION

NONPROVISIONAL APPLICATION TRANSMITTAL RULE §1.53(b)

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

Transmitted herewith for filing under 37 C.F.R. §1.53(b) is the nonprovisional patent application

For (Title): A METHOD AND AN APPARATUS FOR PROCESSING AN AUSCULTATION SIGNAL

By (Inventors): Lars ARKNAÆS-PEDERSEN (Struer, DENMARK)

- ☒ Formal drawings (Figs. 1-10b; eight sheets) are attached.
- ☐ A Declaration and Power of Attorney is filed herewith.
- ☐ An assignment of the invention to _____ is filed herewith.
- ☐ An Information Disclosure Statement is filed herewith.
- ☐ A statement to establish small entity status under 37 C.F.R. §§1.9 and 1.27 is filed herewith.
- ☒ A Preliminary Amendment is filed herewith.
- ☐ Please amend the specification by inserting before the first line the sentence --This nonprovisional application claims the benefit of U.S. Provisional Application No. _____, filed _____.--
- ☒ Priority of foreign application No. 0515/98 filed April 8, 1998 in DENMARK is claimed (35 U.S.C. §119).
- ☒ A certified copy of the above corresponding foreign application(s) is filed herewith.
- ☒ The filing fee is calculated below:

CLAIMS IN THE APPLICATION AFTER ENTRY OF ANY PRELIMINARY AMENDMENT NOTED ABOVE

FOR:	NO. FILED	NO. EXTRA	SMALL ENTITY RATE	SMALL ENTITY FEE		OTHER THAN A SMALL ENTITY RATE	OTHER THAN A SMALL ENTITY FEE
BASIC FEE				$ 380	OR		$ 760
TOTAL CLAIMS	29 - 20	= 9	x 9 =	$	OR	x 18	$ 162
INDEP CLAIMS	3 - 3	= 0	x 39 =	$	OR	x 78	$
☐ MULTIPLE DEPENDENT CLAIMS PRESENTED			+130 =	$	OR	+260	$
			TOTAL	$	OR	TOTAL	$ 922

* If the difference is less than zero, enter "0".

- ☒ Check No. 67355 in the amount of $922.00 to cover the filing fee is attached. Except as otherwise noted herein, the Commissioner is hereby authorized to charge any other fees that may be required to complete this filing, or to credit any overpayment, to Deposit Account No. 15-0461. Two duplicate copies of this sheet are attached.
- ☐ This application is entitled to small entity status. DO NOT charge large entity fees to our Deposit Account.

Respectfully submitted,

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PATENT APPLICATION

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of

Lars ARKNÆS-PEDERSON

Application No.: New U.S. Patent Application

Filed: April 1, 1999 Docket No.: 103176

For: A METHOD AND AN APPARATUS FOR PROCESSING AN AUSCULTATION SIGNAL

PRELIMINARY AMENDMENT

Assistant Commissioner of Patents
Washington, D. C. 20231

Sir:

Prior to initial examination, please amend the above-identified application as follows:

IN THE CLAIMS:

Please amend claims 4, 5, 7-14, 18, 19 and 21-28 as follows:

Claim 4, line 2, change "claims 1-3" to --claim 1--.

Claim 5, line 2, change "claims 1-4" to --claim 1--.

Claim 7, line 2, change "claims 3-5" to --claim 3--.

Claim 8, line 2, change "claims 1-7" to --claim 1--.

Claim 9, line 2, change "claims 1-8" to --claim 1--.

Claim 10, line 2, change "claims 1-9" to --claim 1--.

Claim 11, line 2, change "claims 1-10" to --claim 1--.

Claim 12, line 2, change "claims 1-11" to --claim 1--.

Claim 13, line 2, change "claims 1-12" to --claim 1--.

Claim 14, line 2, change "claims 1-13" to --claim 1--.

Claim 18, line 1, change "claims 15-17" to --claim 15--.

Claim 19, line 1, change "claims 15-19" to --claim 15--.

Claim 21, line 1, change "claims 17-20" to --claim 17--.

Claim 22, line 2, change "claims 15-21" to --claim 15--.

Claim 23, line 1, change "claims 15-22" to --claim 15--.

Claim 24, line 1, change "claims 15-23" to --claim 15--.

Claim 25, line 1, change "claims 15-24" to --claim 15--.

Claim 26, line 1, change "claims 15-25" to --claim 15--.

Claim 27, line 1, change "claims 15-26" to --claim 15--.

Claim 28, line 1, change "claims 15-27" to --claim 15--.

REMARKS

Claims 1-29 are pending. By this Preliminary Amendment, claims 4, 5, 7-14, 18, 19 and 21-28 are amended to eliminate multiple dependencies. Prompt and favorable consideration on the merits is respectfully requested.

Respectfully submitted,

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A method and an apparatus for processing an auscultation signal.

The present invention relates to a method of processing a signal representing an input sound signal, said signal being divided in time into a plurality of signal segments, each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated at least once in said output signal.

Moreover, the present invention relates an apparatus, and in particular to an electronic stethoscope for use in cardiology.

Through the recent years physicians have been provided with an impressive arsenal of instrumentation for the diagnosis of cardiovascular diseases. Such an instrument is the well known stethoscope used to detect sounds originating from the heart and adjacent large vessels. Sound monitoring of the heart, or auscultation in general, is an important aspect in the evaluation of the physical condition of an individual, and is particularly important in the diagnosis of certain pathological conditions which manifest themselves by abnormal sounds.

When using a normal bifurcated stethoscope with binaural earpieces and a bell or diaphragm for receiving the sound signal, it is difficult to distinguish the sound elements in fast beating hearts, e.g. infants, but also when auscultating patients with a 'normal' heart rate, it can be

difficult to observe split heart sounds or a weak murmur located near a primary heart sound.

Today it is possible to process the information residing in the auscultation signal electronically by using knowledge obtained by clinical research. Electronic stethoscopes make it possible to modify the physiological signal, but the approaches are mostly based on changes of the frequency components in the signal, which makes it difficult for a physician, trained in the use of the conventional stethoscopes, to recognize the signal.

This leads to the goal of creating a stethoscope or an apparatus for auscultation in general that makes it easier for the pathologist to distinguish between the different sound elements in even fast heart sounds. Since the pathologists partly base their diagnosis on the heart sounds, it is of great importance that an exact reproduction of the sound elements in the sound signal is performed, meaning that there should be no change in the pitch of the signal and no dissonance should be added as a result of the reproduction algorithm. If either distortion or change of pitch is present, it could lead to a wrong interpretation of the heart sounds, resulting in incorrect diagnoses by the physicians.

US patent No. 4,528,689 discloses the idea of a method for artificially slowing down an analyzed sound signal. It is done by first low-pass filtering the sound signal from the heart and then splitting the signal, which varies cyclically from zero crossing to zero crossing, into a number of cycles, and each cycle is repeated successively. These repetitions of half-periods of a sound representing signal result in a slow version of the original sound having the original pitch.

Consequently, the prior art involves the problem that the resulting signal provides echoes in such a way that a listener may obtain confused results while listening to the generated slow signal. Appearances of echoes might result in a wrong interpretation of some sound elements, and these sound elements are often of vital importance. When identifying a heart disease, this method might lead to incorrect diagnoses. Further the method introduces click sounds at the points where successive cycles are pasted together. Dissonance as click sounds might also lead to disturbance of the auscultation signal and even wrong diagnoses. Moreover, the prior art does not take into account that the auscultation signal can be acquired under disturbed or varying conditions. Apparently, the method has found very little commercial use, if any.

It is an object of the invention to provide a method which will be able to slow down sound signals especially sound signals representing auscultation signals like heart sounds, while still obtaining the original pitch with a minimum of echo in the resulting slowed down signal.

This is achieved, when the method mentioned in the opening paragraph is characterized in that each signal segment is established such that the duration of time of substantially all the signal segments is less than a limit of 50 ms.

Consequently, it is possible for a listener to distinguish between the different sounds of the generated output signal, as the audio signals generated according to the invention are ideally free of any perception of echo or significant distortion. This possibility of eliminating the echo perception for a listener makes the inven-

tion a unique and essential part of any analysis tool for supporting signal analyzing based on subjective recorded and processed audio signals. Using this invention the physician will not be able to distinct between a 'real' auscultation signal and a processed auscultation signal, as the processed signal will be perceived as a 'real' signal.

This feature is of particular importance when speaking about sound signals, which can only be analyzed by means of a subjective analysis performed by a listener. Fields in which the invention will provide important support include evaluation of sound signals emitted from the heart beats. A trained listener, such as a pathologist, will thus have the possibility of getting a full expression of the actual emitted sound, even if the emitted signal is a high-speed signal, such as the one provided by the heart of a child.

In an expedient embodiment the auscultation signal is filtered iteratively by means of an iterative filtering process until the duration of time of substantially all the signal segments is less than the limit. Thereby, the signal processing is capable of adapting itself to a broad spectrum of different auscultation signals.

This is possible in particular when the iterative filtering process is terminated when the filtered signal does not comprise signal segments having a duration of time which is longer than the limit. Thereby, an auscultation signal comprising relatively small signal amplitudes at low frequencies is not filtered too much, which otherwise would cause an excessive number of segments having a short duration of time. An excessive number of segments

having a short duration of time will increase the computational effort required for repeating the segments.

When the limit is less than 40 ms, preferably 30 ms a very preferred embodiment according to the invention is achieved, as pilot tests have turned out to be very successful with respect to e.g. stethoscopes. The embodiment of the invention thus provides no perception of echo even if the signal comprises signal components in a frequency spectrum of a recorded heart sound signal of between approximately 20Hz-2kHz.

In an expedient embodiment the auscultation signal is pre-filtered iteratively by means of a high-pass filter until the duration of time of signal segments is less than the limit. Thereby, a tangible stopping criteria for the iterative filtering process is provided.

When, moreover, the output signal is post-filtered iteratively with a filter having an amplitude transfer function corresponding to the inverse amplitude transfer function of the high-pass filter, the frequency-amplitude response of the pre- and post-filtering process is substantially flat.

In an expedient embodiment the iterative filtering process is terminated when the auscultation signal has been filtered a specified number of times and that an indicator signal indicating termination of the filtering process is provided. Thereby it is possible to select a threshold for a maximum allowable filtering iterations which when reached may generate a warning signal.

Since successive repeating of signal segments having a relatively short duration will result in a poor sound quality at relatively high frequencies the signal seg-

ments having a relatively short duration of time are patched together to form a coherent segment comprising at least three zero-crossings, which coherent segment is repeated at least once.

When the input signal is divided into signal segments in zero crossings excessive high-frequency components are avoided which otherwise can ruin the sound quality.

When the input signal is divided into signal segments such that the gradients of neighboring signal segments of the output signal are substantially equal, and wherein the neighboring signal segments are level-compensated, high-frequency components are reduced to a minimum.

In an expedient embodiment the signal divided segments are multiplied or filtered by means of a window function such that the transitions between neighbouring signal segments are smoothed.

In a simple and preferred embodiment of the invention, the signal segments are reversed with respect to time before the repetition, thus ensuring that the repetitions will have a kind of short duration "backward" masking. It should be noted that the necessary signal processing for obtaining the inverse repetition described above is minimal.

Moreover, the signal segments in the output signal can be mirrored about a time axis in order to further smooth the transitions between neighbouring segments.

In a preferred embodiment the auscultation signal is prefiltered by a high-pass filter such that further zero crossings may be obtained.

Moreover, the invention relates to an apparatus and in particular to a stethoscope.

The invention will be explained more fully below in connection with a preferred embodiment and with reference to the drawing, in which:

fig. 1 illustrates the basic parts of an electronic stethoscope,

fig. 2 shows the half rate algorithm consisting of a filter algorithm, a copy-and-splice (CAS) algorithm and the buffers,

fig. 3a shows the iterative filter process of the filter algorithm and figure 3b, 3c, 3d and 3e illustrates the algorithm performed on heart stepwise,

fig. 4 illustrates a signal with zero crossings and a window function,

fig. 5 illustrates the CAS algorithm when performed on a sample signal of two cycles,

fig. 6 illustrates a heart signal before pre-filtering,

fig. 7 illustrates a heart signal after pre-filtering,

fig. 8 illustrates a half rate heart signal before post-filtering,

fig. 9 illustrates a half rate heart signal after post-filtering, and

fig. 10a illustrates the pre-filter and figure 10b illustrates the post-filter.

Fig. 1 illustrates an electronic stethoscope consisting of a microphone 11 connected to an analog to digital converter 12 from which the output is connected to a memory and central processing unit 13. The memory and central processing unit 13 is connected to a digital to analog converter 14 and the output is connected to a speaker 15. Thereby an auscultation signal is acquired, processed, and reproduced as a sound signal.

In use, the physician places the microphone 11, which may be in the shape of a bell, on the patient´s chest and the sound is recorded and processed in the processing unit 13. It is possible to hear the processed signal by using the speaker or speakers 15 connected to the digital to analog converter 14.

Fig. 2 illustrates the half rate algorithm performed by the central processing unit and memory 13. This algorithm consists of two parts - a filter algorithm 21 and a copy-and-splice (CAS) algorithm 22. The recorded data is placed in the input work buffer 23, and then filtered using an iterative filter 24 and 24′. The CAS algorithm 22 is performed between the filters 24 and 24′. The CAS algorithm consists of a zero crossing locator 25, a window function 26 and a copy and splice function 27. The algorithm halves the rate of the sound, resulting in a doubling of the length of the sound signal, whereby the signal has twice the original duration in the output work buffer 28.

The iterative filter algorithm including the pre-filter 35 and the post-filter 37 is shown in figure 3a. The input signal is pre-filtered. This is done in order to am-

plify the high frequency signal elements and attenuate the lower frequencies. In order to reduce the processor power needed by the algorithm, the algorithm is performed on the sound signal part by part. In a preferred embodiment it is run on parts with a 10 second duration. First, the time period 32 is singled out of the recorded sound signal 31, secondly an algorithm 33 determines the maximum cycle time. In 34 the algorithm checks whether the cycle time is above a predetermined value T_{max}. If it is above the predetermined value, the signal part is filtered using a high-pass filter 35, and this step is repeated until the cycle time is below the predetermined value. Then, the CAS algorithm is executed in 36, in the described embodiment the algorithm doubles the length of the sound signal. Finally, the signal part is post-filtered the same number of times as it was pre-filtered using a low pass filter 37 which has an inverse transfer function with respect to the pre-filter 35. The post-filtering 37 amplifies the low frequencies (long cycles) in the same way as they were attenuated in the pre-filter 35 in order to ensure a flat frequency response from input to output. To avoid echo, the value T_{max} should be chosen according to the time constant of the ear, which is the response time for the human ear after hearing a first sound.

To illustrate the effect of the iterative filter, a signal is shown before filtering in figure 3b. Then the signal is shown after one filtering in figure 3c, followed by the signal after filtering twice. Finally, the signal is shown after being filtered three times resulting in an extra zero crossing in the time interval between 0,6s and 0.65s.

The CAS algorithm shown in figure 2 will be described in detail below. A sample signal 43 is shown in figure 4. The zero crossing locator 25 in figure 2 locates the negative to positive transitions 41 (zero crossings) in the filtered input signal. This means that the boundaries of all cycles in the signal are located. These locations will be used by the window function 26 shown in figure 2. The window function 26 is used to prevent click sounds from occurring when succeeding cycles are patched together, the start and end portion of each cycle are smoothed (faded in/out). The window 42 will generate signal portions that are a bit longer than those of the cycle itself (zero crossing to zero crossing), this is done to enable smooth overlapping sections, from one cycle to the next. In the preferred embodiment the amplitude (weight) of the window 42 at its centre equals 1.0, and the weight at the zero crossings 41 equals 0.5. This results in the cycle after cutting 44 being a bit longer than from zero crossing to zero crossing, providing smooth transitions between succeeding cycles.

An example of how the output signal of a half-rate signal is made by using the copy and splice process 27 from figure 2 is shown in figure 5. A sample input signal after pre-filtering 51 consists of two cycles 52 and 53. The cycle 52 is cut from the sample signal 51, using the window 42 shown in figure 4, providing the signal 54. It is seen that the window described above used on the cycle 52 results in signal 54 with longer duration than the identified cycle 52. The signal 54 is then copied and shifted in time resulting in the signal 55, which succeeds the signal 54 with overlapping zero crossing. In a preferred embodiment as shown in figure 5, the copy 55 is mirrored in both the horizontal axis and the vertical i.e. re-

versed in time and mirrored about a time axis. Tests have shown that this results in a minimum chance of echo perception. The similar is done to the cycle 53, first the cycle is cut from the signal 51 using the window 42 providing the signal 56. Then the signal 56 is copied and mirrored providing the signal 57. The signals 54, 55, 56 and 57 are added causing a reduction of the rate of the signal 51 by 50%. The original pitch is obtained, and performing this process on only fast cycles, the listener will not experience any echoes.

This method used on a heart signal is shown in the following.

Fig. 6 shows a heart signal before filtering with the iterative high-pass pre-filter 24 from figure 2. It is obvious that the signal includes some slow cycles.

Fig. 7 shows the signal after filtering with the high pass filter 24, and it is obvious that the slow cycles have been attenuated resulting in only fast cycles.

Fig. 8 shows the pre-filtered heart signal as it is before post-filtering. The length of the signal has been doubled using the CAS algorithm.

Fig. 9 shows the output signal after post-filtering. The signal has been post-filtered the same number of times as it was pre-filtered. The rate of the original sound signal has now been halved and the physician listening to this halved version will not be able to perceive any echo.

Fig. 10 shows the pre-filter and the post-filter, which in combination have a flat frequency response.

It should be noted that in this preferred embodiment of the invention the rate of the signal has been halved but it is also possible to reduce the speed by another fraction. This reduction depends on how many times the cycles are repeated. In this embodiment cycles were used as the segments to be copied, but other methods could also be used to define a segment. Though it is advantageously to use segments that makes it possible to get a smooth transition between neighboring segments.

Further, it should be noted that the number of iterations in the iterative filter depends on the auscultation signal in question. Typically, an auscultation signal acquired from the breast-case of an adult breathing normally requires 2-3 iterations, whereas an auscultation signal acquired from the lungs of an adult breathing normally requires 1-2 iterations. A maximum number of iterations is specified to 5 iterations.

Still further, it should be noted that iterative filter can use another stopping criteria e.g. Tmax < 25 ms and/or that a specified maximum number of iterations has been reached. When a maximum number of iterations has been reached a signal can be provided which can be used to warn a user and/or terminate processing of an actual auscultation signal. The stopping criteria depends on the iterative filter itself and any filtering of the auscultation signal prior to the iterative filtering.

Experiments has shown that when a signal input to the CAS algorithm comprises many segments having a short duration of time the sound quality of the reproduced sound signal is ruined. Therefore, the iterative filter having a stopping criteria ensures that there is not generated an ex-

cessive number of segments having a short duration of time.

However the CAS algorithm can be adapted to handle a succession of segments having a short duration of time (e.g. less than 4-5 ms) if such a succession occurs.

When a succession of segments having a short duration of time is detected, the CAS algorithm patch the succession of segments together to form a coherent segment, which coherent segment is repeated a specified number of times corresponding to the rate at which the auscultation signal shall be slowed down. It should be noted that the coherent segment corresponds to a given part of the auscultation signal wherein there is a number of zero-crossings.

Another way to obtain additional zero-crossings in the auscultation signal would be to process the signal through a linear prediction-error filter. In this case, the function of the prediction-error filter would be to whiten the signal, so further zero-crossings would occur. The coefficients used in the all-zero linear prediction-error filter can be found, using well known analysis methods like "Levinson-Durbin recursion", "Burg algorithm" or others. The same coefficients are used in a all-pole output synthesizer filter, to which the output signal from the CAS algorithm is applied. The output synthesizer filter is used to ensure a flat frequency response from input to output.

The invention can be embodied as a part of a stethoscope or any other instrument or apparatus. The physical embodiment of a stethoscope according to the invention can be embodied e.g. as shown in fig. 1 of US patent

4,528,689. Additionally, a stethoscope according to the invention may comprise a readout of a heart rate calculated from the auscultation signal.

In a preferred embodiment segments are defined as a part of a signal from a zero-crossing having a positive or negative gradient to a zero-crossing having a positive or negative gradient, respectively.

Alternatively, the invention may be embodied as a computer program or a part of a computer program, which may be loaded into the memory of a computer and executed therefrom. The computer program may be distributed by means of any data storage or data transmission medium. The storage media can be magnetic tape, optical disc, compact disc (CD or CD-ROM), mini-disc, hard disk, floppy disk, ferroelectric memory, electrically erasable programmable read only memory (EEPROM), flash memory, EPROM, read only memory (ROM), static random access memory (SRAM), dynamic random access memory (DRAM), ferromagnetic memory, optical storage, charge coupled devices, smart cards, etc. The transmission medium can be a network, e.g. a local area network (LAN), a wide area network (WAN), or any combination thereof, e.g. the Internet. The network may comprise wire and wire-less communication links. Via the network a software embodiment (i.e. a program) of the invention, or a part thereof, may be distributed by transferring a program via the network.

CLAIMS

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1. A method of processing an auscultation signal, said auscultation signal being divided into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed such that at least one, preferably all signal segments are repeated at least once in said output signal,

characterized in that

each signal segment is established such that the duration of time of substantially all the signal segments is less than a limit of 50 ms.

2. A method of processing an auscultation signal according to claim 1, characterized in that the auscultation signal is filtered iteratively by means of an iterative filtering process until the duration of time of substantially all the signal segments is less than the limit.

3. A method of processing an auscultation signal according to claim 2, characterized in that the iterative filtering process is terminated when the filtered signal does not comprise signal segments having a duration of time which is longer than the limit.

4. A method of processing an auscultation signal according to claims 1-3, characterized in that the limit is less than 40 ms, preferably 30 ms.

5. A method of processing an auscultation signal according to claims 1-4, characterized in that the auscultation signal is pre-filtered iteratively by means of a high-pass filter until the duration of time of signal segments is less than the limit.

6. A method of processing an auscultation signal according to claim 5, characterized in that the output signal is post-filtered iteratively with a filter having an transfer function corresponding to the inverse amplitude transfer function of the high-pass filter.

7. A method of processing an auscultation signal according to claims 3-5, characterized in that the iterative filtering process is terminated when the auscultation signal has been filtered a specified number of times and that an indicator signal indicating termination of the filtering process is provided.

8. A method of processing an auscultation signal according to claims 1-7, characterized in that signal segments having a relatively short duration of time are patched together to form a coherent segment comprising at least three zero-crossings, which coherent segment is repeated at least once.

9. A method of processing an auscultation signal, according to claims 1-8, characterized in that the auscultation signal is divided into signal segments in zero crossings.

10. A method of processing an auscultation signal according to claims 1-9, characterized in that the auscultation signal is divided into signal segments such that the gradients of neighboring signal segments of the output signal are substantially equal, and wherein the neighboring signal segments are level-compensated.

11. A method of processing an auscultation signal according to claims 1-10, characterized in that the signal divided segments are multiplied or filtered by means of a window function such that the transitions between neighbouring signal segments are smoothed.

12. A method of processing an auscultation signal according to claims 1-11, characterized in that signal segments in the output signal are reversed in time.

13. A method of processing an auscultation signal according to claims 1-12, characterized in that signal segments in the output signal are mirrored about a time axis.

14. A method of processing an auscultation signal according to claims 1-13, characterized in that the auscultation signal is pre-filtered by a high-pass filter such that further zero crossings may be obtained.

15. An apparatus for processing an auscultation signal, the apparatus comprising signal processing means for dividing the auscultation signal into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed such that at least one, preferably all signal segments are repeated at least once in said output signal

characterized in that

said apparatus further comprises means for establishing each signal segment such that the duration of time of substantially all the signal segments is less than a limit of 50 ms.

16. An apparatus according to claim 15, characterized in that the apparatus comprises means for filtering the auscultation signal iteratively by means of an iterative filtering means until the duration of time of substantially all the signal segments is less than the limit.

17. An apparatus according to claim 16, characterized in that the iterative filtering means is interrupted when the filtered signal does not comprise signal segments having a duration of time which is longer than the limit.

18. An apparatus according to claims 15-17, characterized in that the limit is less than 40 ms, preferably 30 ms.

19. An apparatus according to claims 15-19, characterized in that the apparatus comprises a high-pass filter for pre-filtering the auscultation signal iteratively until the duration of time of signal segments is less than the limit.

20. An apparatus according to claim 19, characterized in that apparatus comprises a filter having an amplitude transfer function corresponding to the inverse amplitude transfer function of the high-pass filter, for post-filtering the auscultation signal

21. An apparatus according to claims 17-20, characterized in that the iterative filtering means is interrupted when the auscultation signal has been filtered a specified number of times and that an indicator signal indicating termination of the filtering process is provided.

22. A method of processing an auscultation signal according to claims 15-21, characterized in that signal seg-

ments having a relatively short duration of time are patched together to form a coherent segment comprising at least three zero-crossings, which coherent segment is repeated at least once.

23. An apparatus according to claims 15-22, characterized in that the apparatus comprises means for dividing the auscultation signal into signal segments in zero crossings.

24. An apparatus according to claims 15-23, characterized in that the apparatus comprises means for dividing the auscultation signal into signal segments such that the gradients of neighboring signal segments of the output signal are substantially equal, and wherein the neighboring signal segments are level-compensated.

25. An apparatus according to claims 15-24, characterized in that the apparatus comprises means for multiplying or filtering the signal divided segments by a window function such that the transitions between neighbouring signal segments are smoothed.

26. An apparatus according to claims 15-25, characterized in that the apparatus comprises means for reversing the signal segments in the output signal in time.

27. An apparatus according to claims 15-26, characterized in that the apparatus comprises means for mirroring the signal segments in the output signal about a time axis.

28. An apparatus according to claims 15-27, characterized in that the apparatus comprises a high-pass filter for pre-filtering the auscultation signal such that further zero crossings may be obtained.

29. An electronic stethoscope comprising at least one input transducer and at least one output transducer, said stethoscope comprising signal processing means for dividing said input signal in time into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments, said signal segments being processed in such a way that at least one, preferably all signal segments are repeated at least once in said output signal

characterized in that

said stethoscope further comprises means for establishing each signal segment such that the duration of time of substantially all the signal segments is less than 50 ms and said stethoscope further comprising means for reproducing said output signal by means of said output transducers.

ABSTRACT

The invention relates to a method of processing a signal representing an input sound signal, said signal being divided in time into a plurality of signal segments each having an individual duration of time, said signal segments being processed into an output signal of successive signal segments in such a way that at least one, preferably all signal segments are repeated immediately and successively at least once in said output signal, wherein each signal segment is established in such a way that the duration of time of a majority of, preferably all the signal segments is less than 60 ms.

Thus, according to the invention, a sound signal can be reduced in speed, by doubling the number of short cycles.

Fig. 2 should be published.

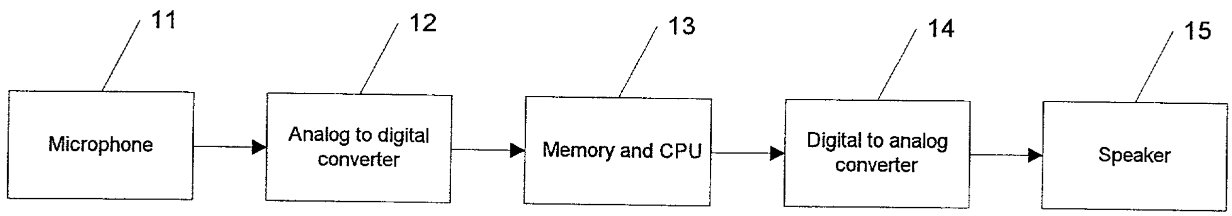

Figure 1

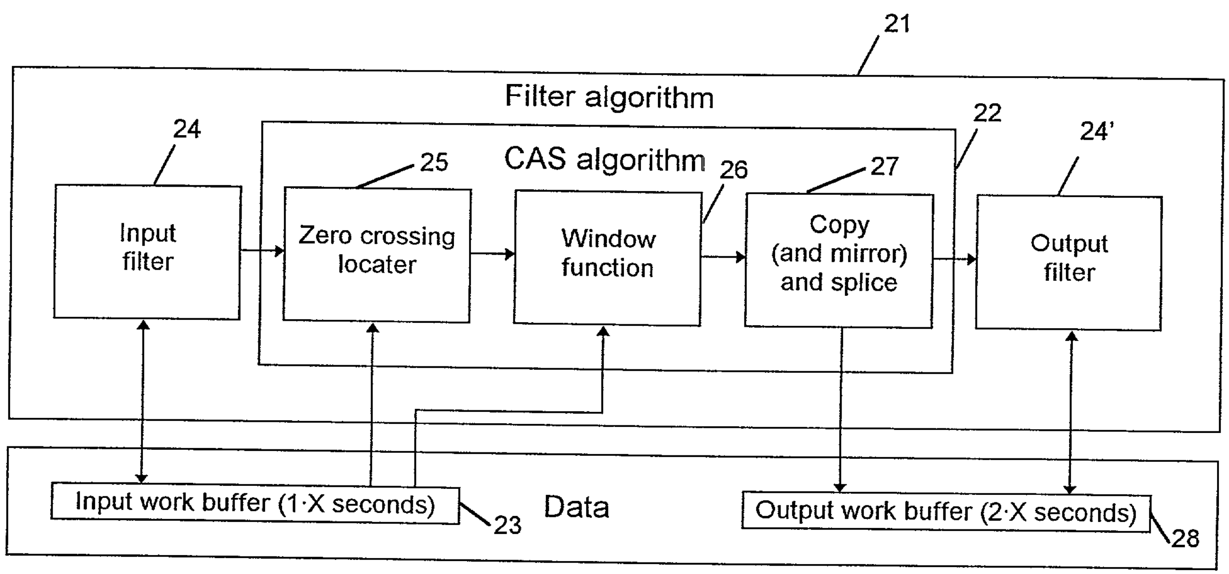

Figure 2

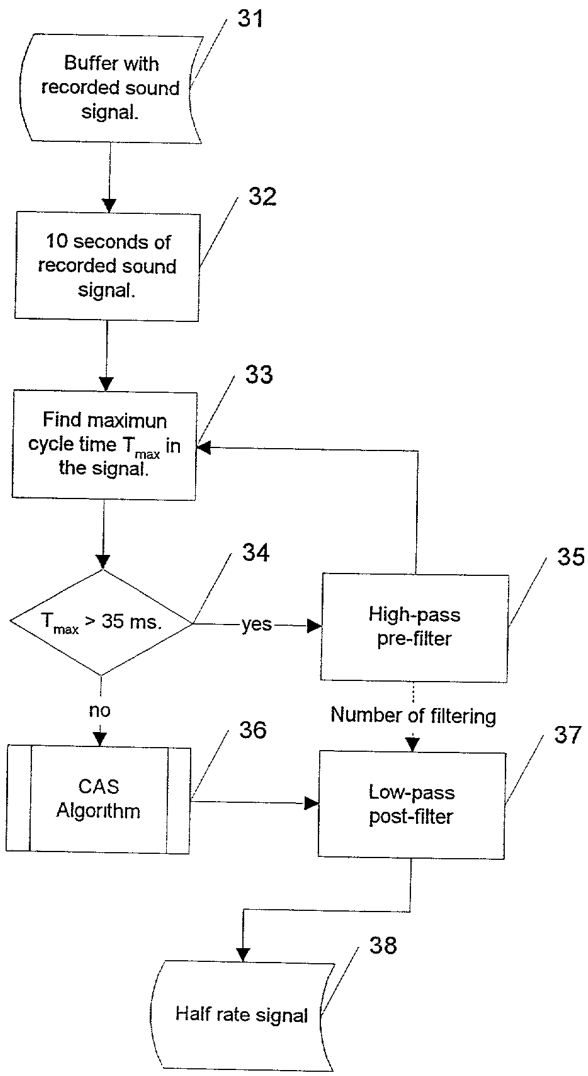
31
Buffer with recorded sound signal.
32
10 seconds of recorded sound signal.
33
Find maximun cycle time T_{max} in the signal.
34
T_{max} > 35 ms.
yes
no
35
High-pass pre-filter
Number of filtering
36
CAS Algorithm
37
Low-pass post-filter
38
Half rate signal

Figure 3a

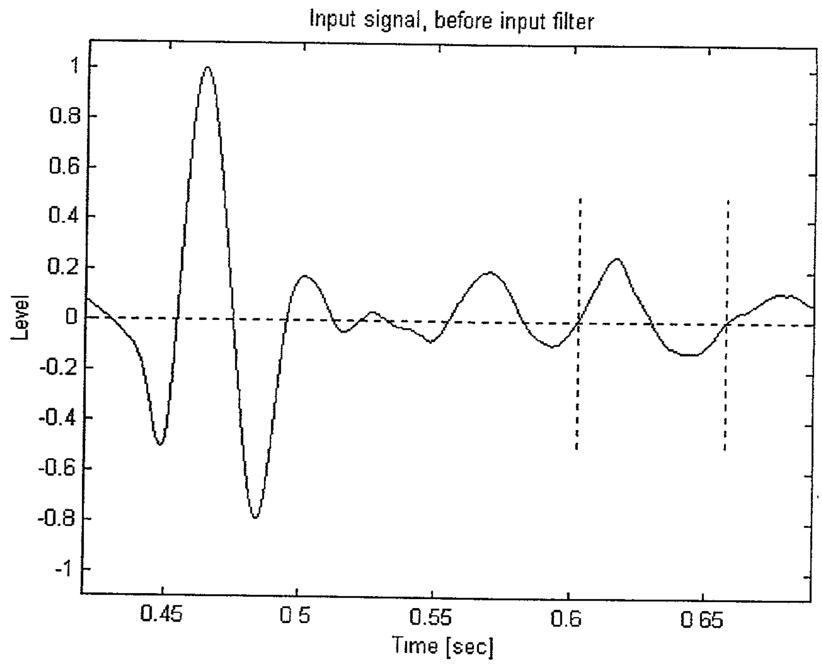

Figure 3b

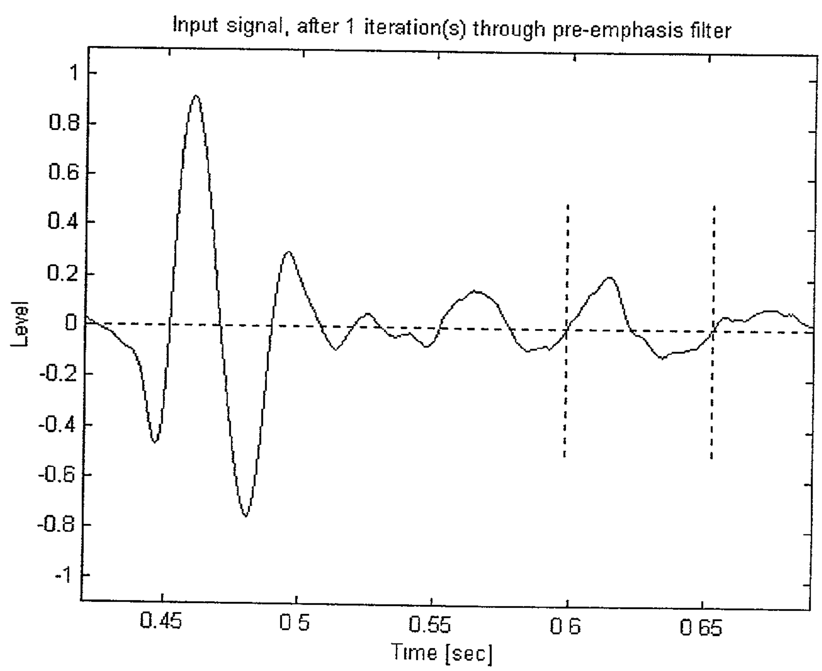

Figure 3c

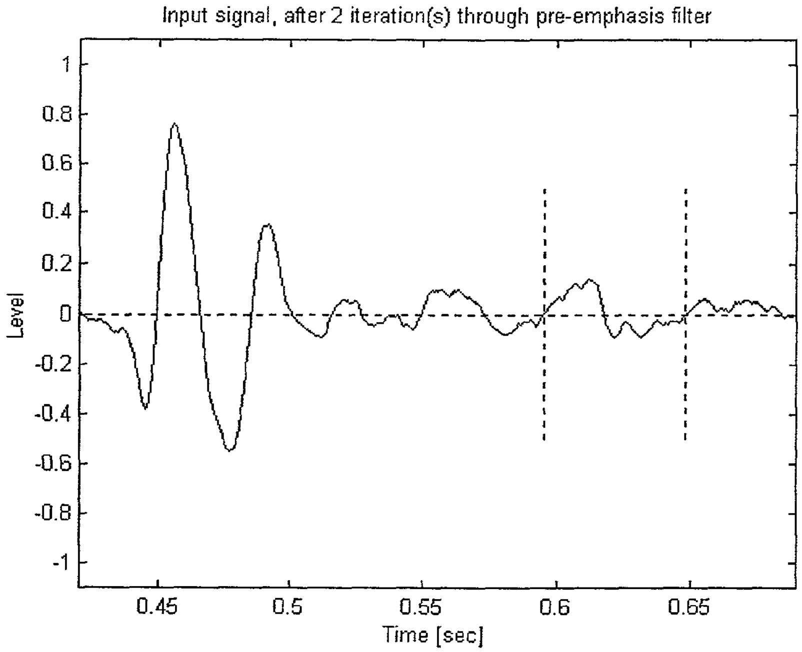

Figure 3d

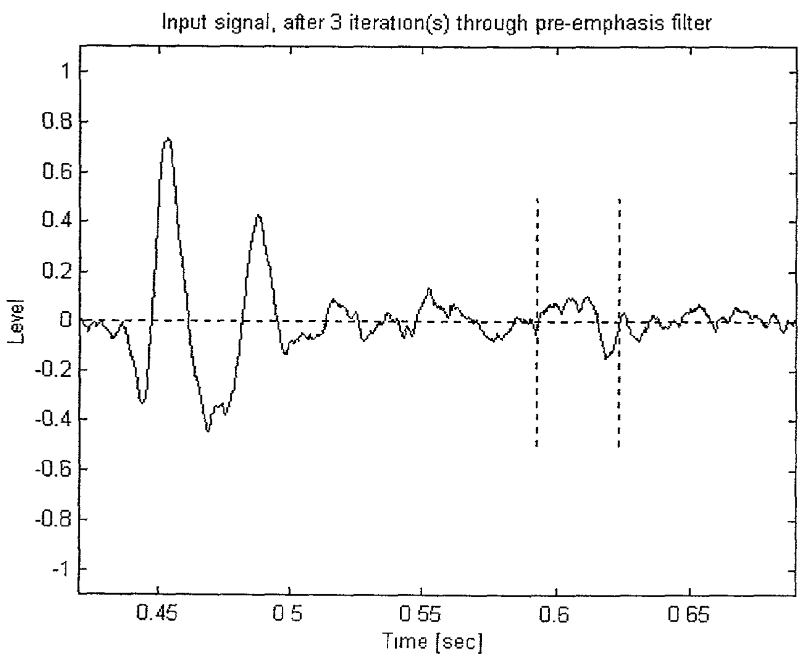

Figure 3e

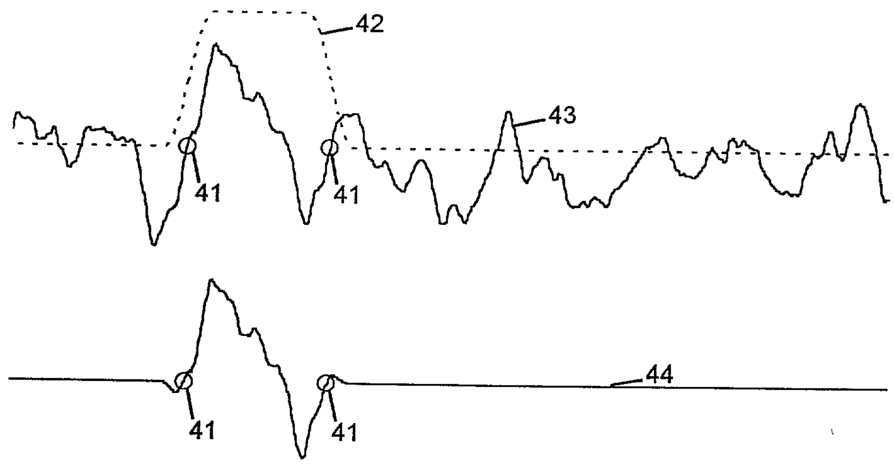

Figure 4

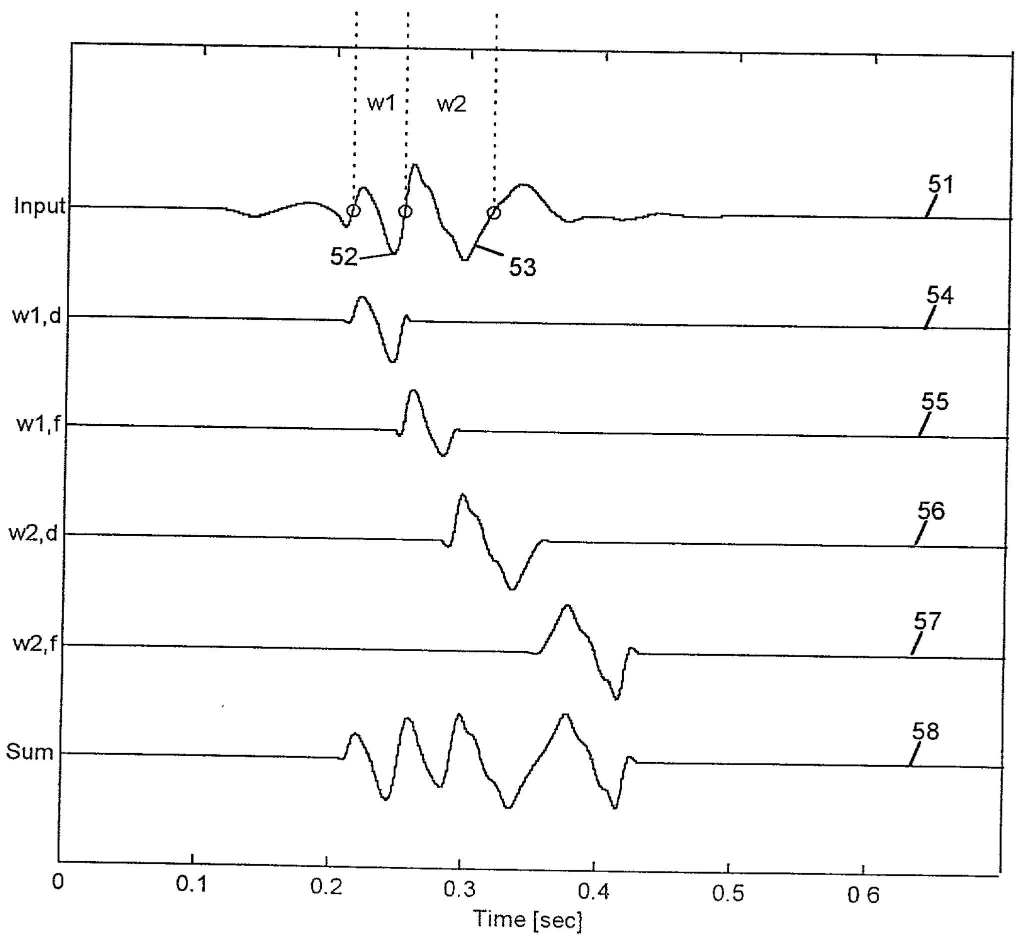

Figure 5

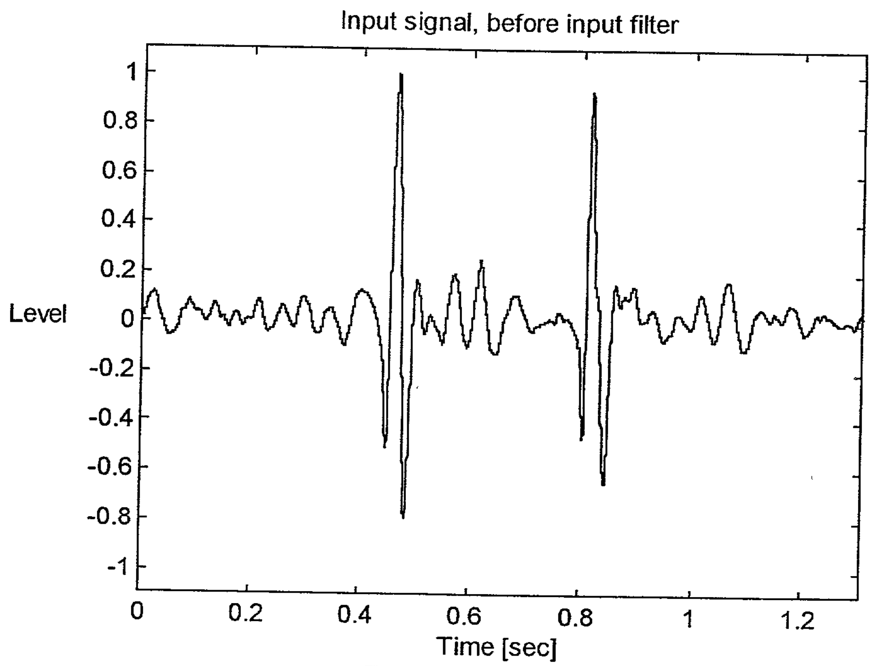

Figure 6

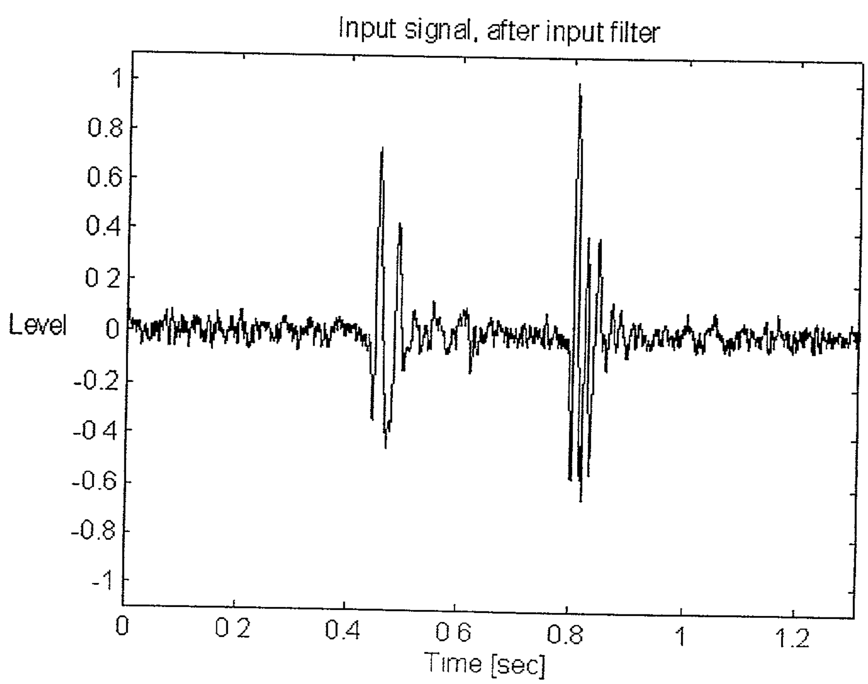

Figure 7

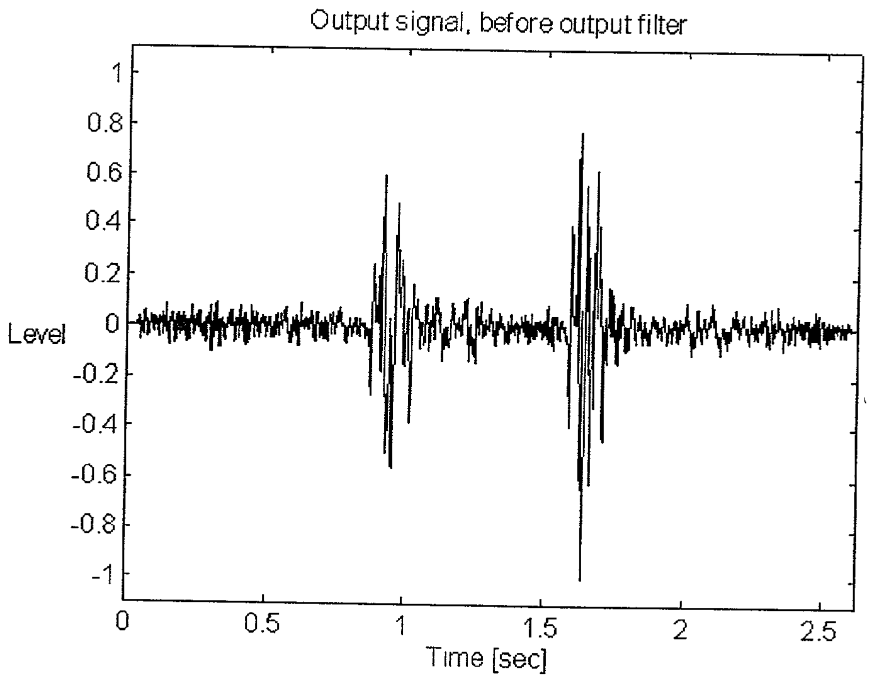

Figure 8

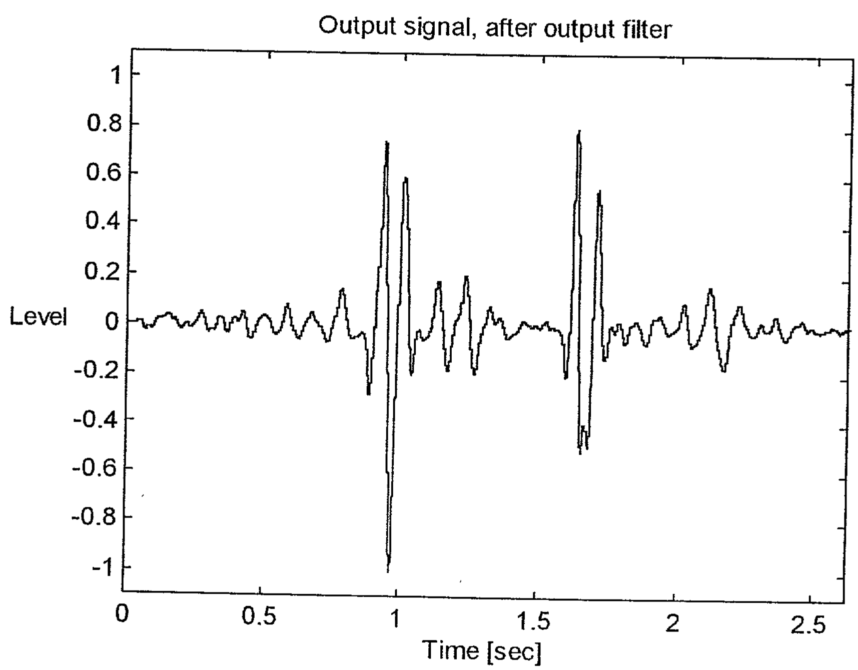

Figure 9

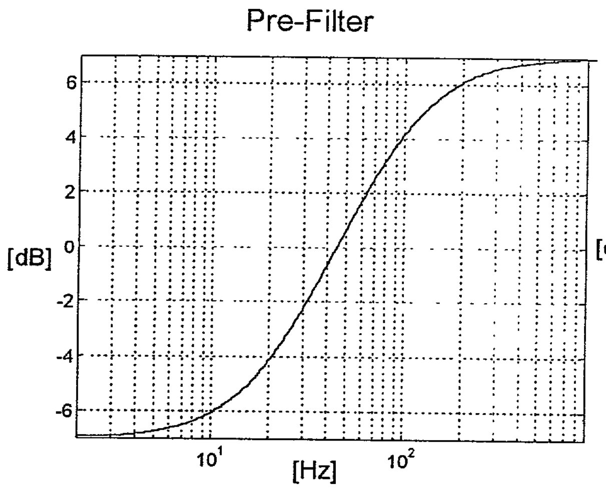

Figure 10a

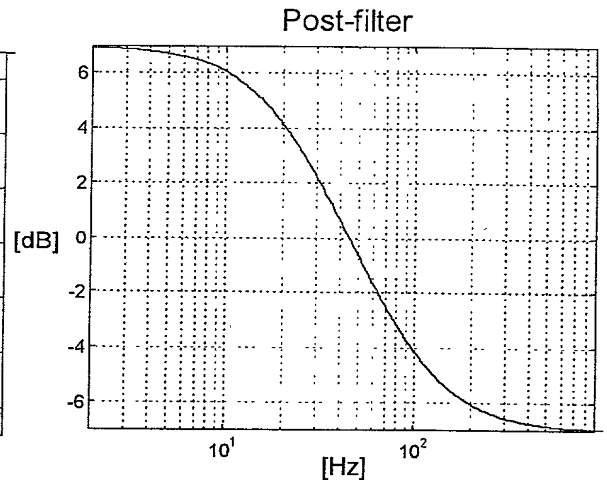

Figure 10b